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PROVISIONAL APPLICATION FOR PATENT COVER SHEET

This is a request for filing a PROVISIONAL APPLICATION FOR PATENT under 37 CFR 1.53(c).

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[X] *Additional inventors are being named on the* 1 *separately numbered sheets attached hereto.*

TITLE OF THE INVENTION (280 characters max)

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ENCLOSED APPLICATION PARTS (*check all that apply*)

[ ] Specification (includes drawings) *Number of Pages* 63 — [ ] CD(s), Number [ ]

[ ] Drawing(s) *Number of sheets* [ ] — [ ] Other (specify) [ ]

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METHOD OF PAYMENT OF FILING FEES FOR THIS PROVISIONAL APPLICATION FOR PATENT (check one)

[X] Applicant claims small entity status. See 37 CFR 1.27.

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The invention was made by an agency of the United States Government or under a contract with an agency of the United States Government.

[X] No.

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Respectfully submitted,

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Date March 4, 2004

REGISTRATION NO. 24,156

(if appropriate)

Docket Number: 206,469

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Number 2 of 2

FMC Product Specification

Index:

1.Introduction

1.1 Overview

This Product Specification describes the application and functionality of FMC platform. The FMC Platform is intended to facilitate a full range of IN services to fixed (PSTN & IP) subscribers and to allowing termination and origination onto the PSTN and other PLMN.
The most important advantage of this platform is the ability To register PSTN & IP subscriber on the GSM HLR. This ability is achieved due to the MAP interface towards the GSM system.

1.2 Scope

The document includes the following information:

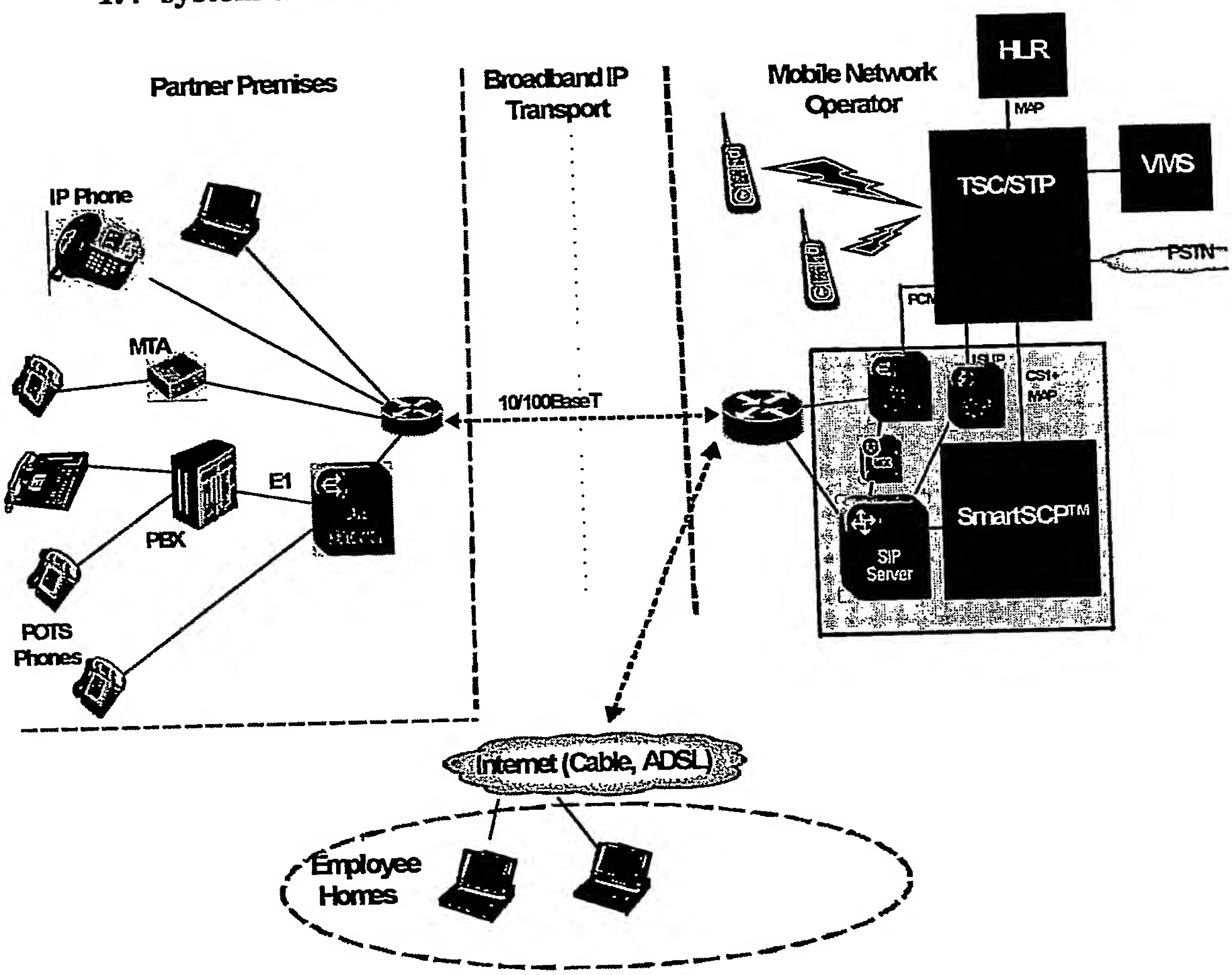
- FMC Specification.
- Functional Description.
- Fraud Management and Financial Audits.
- Operation and Maintenance.
- Other Specification.

1.3 Abbreviations

The terms and abbreviations in this document are explained below.

	Description
CDR	Call Data Record
CS1	Capability Set 1
DB	Data Base
EP	Sip End point User
GW	SIP GW with digital E1 interface and SS7 ISUP protocol
HLR	Home location register
IN	Intelligent Network
INAP	Intelligent Network Application Protocol
ISD	Insert subscriber data (MAP operation)
MAP	Mobile application part
MSISDN	Mobile Station ISDN
PBX	Private branch exchange
PLMN	Public Land Mobile Network
SCCP	Signaling Connection Control Part
SCP	Service control point
SIP	Session interface protocol
SNMP	Simple Network Management Protocol
SS7	Signaling System 7
SSW	Soft Switch
TCAP	Transaction Capabilities Application Part
TCP/IP	Transmission Control Protocol / Internet Protocol
VLR	Visitor location register
VoIP	Voice over IP

1.4 system architecture



2.FMC specification

2.1 Overall objective

- SmartSCP™
- SmartVLR™
- Soft Switch/SIP Server
- Media Gateway
- HLR

2.2 Platform Architecture

The SmartVLR™ will perform VLR functionality using MAP interface to interact with GSM HLR.

The system handles requests from two sources:

1. SIP Server
   Authentication (Registration/Deregistration)
   Call Handling
2. HLR
   Routing Information

The Interface between SmartVLR™ and SIP server is based on TCP/IP connection.

Sip server will request Authentication information when SIP users try to registrate.

Sip server will request user registration and deregistration.

For each request SmartVLR™ will respond will relevant information discuss in 2.3 section.

The Interface between SmartSCP™ and SIP server is based on TCP/IP connection.

Sip server will request instruction for call handling to give them full IN services

The interface between SmartVLR™ and HLR is based on SS7 connection.

In case call from GSM subscriber to SIP subscriber HLR will request Routing information from SmartVLR™

S M A R T

2.3 Interfaces Description
2.3.1 Introduction

The platform provides two main interfaces
1. SmartVLR™ /SmartSCP™ – Sip server interface (through TCP/IP connection)
2. HLR – SmartVLR™ interface (through SS7 connection)

Parameter description:

- Sip user name – user name of Sip subscriber
- Password – password match to Sip user
- Sip URI – URI address of Sip user (like : "sip: 97254160272@orange.co.il")
- Result – used in response messages and have the following values
  - E_RESULT_UNDEFINED = -1
  - E_RESULT_OK = 0
  - E_RESULT_GENERAL_ERROR = 1
  - E_RESULT_INVALID_IP = 2
  - E_RESULT_USER_NOT_FOUND = 3
  - E_RESULT_USER_DISABLED = 4
  - E_RESULT_DEST_DISABLED = 5

- Called – Called party address (format: SIP URI or MSISDN)
- Calling – Calling party address (format: SIP URI)

2.3.2 SmartVLR™ <-> Sip Server Interface

Following table describes SmartVLR™ and SIP Server Interface

Origination	Destination	Message	Parameters	Description
Sip Server	SmartVLR TM	Get Auth Info	1. Sip User name	This message use to request password of SIP user
SmartVLR TM	Sip Server	Get Auth Info Response	1. Result 2. Password	This message is response to Get Auth Info message.
Sip Server	SmartVLR TM	User Attach	1. Sip URI 2. IP Address	This message use to instruct SmartVLR™ to registerate Sip subscriber in HLR
SmartVLR TM	Sip Server	User Attach Response	1. Result	This message is response to User Attach message.
Sip Server	SmartVLR	User Detach	1. Sip URI	This message is used to

	TM			instruct SmartVLR™ to deregistrate Sip subscriber in HLR
SmartVLR TM	Sip Server	User Detach Response	1. Result	This message is used to response User Detach message
SmartVLR TM	Sip Server	Network Detach	1. Sip URI	This message is used to instruct SIP server to

2.3.3 SmartSCP™ <-> Sip Server Interface

Following table describes SmartSCP™ and SIP Server Interface

Sip Server	SmartSCP™	Call Arrive	1. called 2. Calling 3. IP Address	This message is used to request SmartSCP™ instruction for call handling.
SmartSCP™	Sip Server	Connect	1. Called 2. Calling 3. Result	This message is response to Call Arrive message. It used to instruct Sip Server to connect the call to Sip network
SmartSCP™	Sip Server	Connect GSM	1. Called 2. Calling 3. Result	This message is response to Call Arrive message. It used to instruct Sip Server to connect the call to GSM network.

2.3.4 SmartVLR™ <-> HLR Interface

Following table describes SmartVLR™ and HLR Interface.
Note that all services are confirmed services

SmartVLR™	HLR	Update Location	This service is used by the VLR to update the location information stored in the HLR.
SmartVLR™	HLR	**PURGE MS**	This service is used between the VLR and the HLR to cause the HLR to mark its data for an MS so that any request for routing information for a mobile terminated call or a mobile terminated short message will be treated as if the MS is not reachable.
HLR	SmartVLR™	**PROVIDE ROAMING NUMBER**	This service is used between the HLR and VLR. The service is invoked by the HLR to request a VLR to send back a roaming number to enable the HLR to instruct the GMSC to route an incoming call to the called MS.
HLR	SmartVLR™	**CANCEL LOCATION**	This service is used between HLR and VLR to delete a subscriber record from the VLR. It may be invoked automatically when an MS moves from one VLR area to another, to remove the subscriber record from the old VLR, or by the HLR operator to enforce a location updating from the VLR to the HLR.
HLR	SmartVLR™	**INSERT SUBSCRIBER DATA**	This service is used by an HLR to update a VLR with certain subscriber data in the following occasions: - the operator has changed the subscription of one or more supplementary services, basic services or data of a subscriber. - the operator has applied, changed or removed Operator Determined Barring; - the subscriber has changed data concerning one or more supplementary services by using a subscriber procedure; - the HLR provides the VLR with subscriber parameters at location updating of a subscriber or at restoration. In this case, this service is used to indicate

			explicitly that a supplementary service is not provisioned, if the supplementary service specification requires it. The only supplementary services which have this requirement are the CLIR and COLR services.
HLR	SmartVLR™	**DELETE SUBSCRIBER DATA**	This service is used by an HLR to remove certain subscriber data from a VLR if the subscription of one or more supplementary services or basic services is withdrawn.

2.4 Database structure

2.4.1 FMCDB Database

SmartVLR™ will implement internal database in PHASE 1,2 .

The database will include following tables:

1) TBL_SUBS_INFO
2) TBL_TELESERVICE
3) TBL_SS_DATA
4) TBL_Forwarding
5) TBL_BEARER_SERVICE_CODES

2.4.1.1 TBL_SUBS_INFO

This table contains online and constant information of subscriber

UserID	Table Key
IMSI	IMSI of Sip user (uniqe ID in the GSM network)
MSISDN	The subscriber's identify number in the network.
SIP_URI	SIP URI address of Sip subscriber
UserAuthName	Subscriber User name
Password	Subscriber password
MSRN_ID	
IP_ADDRESS	IP Address Sip subscriber register from
HLRNumber	The HLR Number Sip subscriber register to
LastPoolingTime	Last Time subscriber send REGISTER message.
Expiration	The period of time between each REGISTER message.

2.4.1.2 TBL_ TELESERVICE

This table contains Teleservices subscriber allowed to use .
The table contains following fields:

- SUB_ID (Uniqe)
- TELEPHONY
- SMS_MTPP
- SMS_MOPP
- FAXGRP3_ALTSPEECH
- AUTOFAX_GRP3
- FAXGRP4
- VOICE_GRP
- VOICE_BROAD
- TSD1
- TSD2
- TSD3

SMART

- TSD4
- TSD5
- TSD6
- TSD7
- TSD8
- TSD9
- TSDA
- TSDB
- TSDC
- TSDD
- TSDE
- TSDF

2.4.1.3 TBL_SS_DATA

This table contains SS_DATA defined for specific subscriber.
The table contains following fields:

Sub_ID	Uniqe
SSC_CLIP	Calling line identification presentation
SSC_CLIR	Calling line identification restriction
SSC_COLP	Connected line identification presentation
SSC_COLR	Connected line identification restriction
SSC_CW	Call waiting
SSC_HOLD	Call hold
SSC_MULTIPTY	Multi party service
SSC_BAOC	Barring of all outgoing calls
SSC_BOIC	Barring of all outgoing international calls
SSC_BOIC_EX_HC	Barring of all outgoing international calls except those directed to the home PLMN country
SSC_BAIC	Barring of all incoming calls
SSC_BICROAM	Barring of all incoming calls when roaming outside the home PLMN country

2.4.1.4 TBL_Forwarding

This tables contains forwarding information of subscriber
The table contains the following fields:

Sub_ID	Uniqe
ForwardingType	One of the following: • Call Forward Unconditional • Call Forward on MS busy • Call Forward on MS not reply • Call Forward on MS not reachable
Number	Forward Number
SubAddress	Forward SubAddress
BasicServiceType	
BasicService	
NoRepTime	No Answer time in case of Call Forward on MS not reply

2.4.1.5 TBL_BEARER_SERVICE_CODES

This tables contains Bearer service of data and not has influence on SIP subscribers

2.4.1.6 TBL_ODB

This Table contains information about ODB (Operator determined barring) for subscriber. The table contains the following fields:

SUB ID	Uniqe
ITOGCB	International Out Going Calls Barred
ITOGCNotHPLMNCB	International Out Going Calls Not To HPLMN Country Barred
IZOGCB	Interzonal Out Going Calls Barred
IZOGCNotHPLMNCB	Interzonal Out Going Calls Not To HPLMN Country Barred
IZOGCAndITOGCNotHPLMNCB	Interzonal Out Going Calls And International Out Going Calls Not To HPLMN Country Barred
PRInfoOGCB	Premium Rate Information Out Going Calls Barred
PREnterOGCB	Premium Rate Entertainement Out Going Calls Barred
SSAccessB	Ss Access Barred
ALLECTB	All ECT Barred
CHARGEECTB	Chargeable ECT Barred
ITECTB	International ECT Barred
IZECTB	Interzonal ECT Barred
DCHARGEECTB	Doubly Chargeable ECT Barred
MECTB	Multiple ECT Barred
ALLPOServicesB	All Packet Oriented Services Barred
RAToHPLMNAPB	Roamer Access To HPLMN AP Barred
RAToVPLMNAPB	Roamer Access To VPLMN AP Barred
ROPLMNOGCB	Roaming Outside PLMN Out Going Calls Barred
ALLICCB	All in Coming Calls Barred
ROPLMNICCB	Roaming Outside PLMN In Coming Calls Barred
ROPLMNICICCB	Roaming Outside PLMN Country In Coming Calls Barred
ROPLMNB	Roaming Outside PLMN Barred
ROPLMNCB	Roaming Outside PLMN Country Barred
REGALLCFB	Registration All CF Barred
REGCFNotHPLMNB	Registration CF Not To HPLMN Barred
REGIZCFB	Registration Interzonal CF Barred
REGIZCFNotHPLMNB	Registration Interzonal CF Not To HPLMN Barred
REGITCFB	Registration International CF Barred

3.Functional Description

3.1 Introduction

This section describes the functionality of the Inter Platform using flow diagrams, in order to illustrate the requests processing involved with the platform.

There are two main process type involved with the platform

- Registration/Deregistration
- Call Initiation
  - Mobile To Fix
  - Fix to Mobile
  - Fix to Fix
- HLR VLR Interface

3.2 Registration

3.2.1 Subscription & Registration Concept:

Each endpoint should be registered to be able to initiate a call.
The registered endpoints are stored on the SmartVLR™.

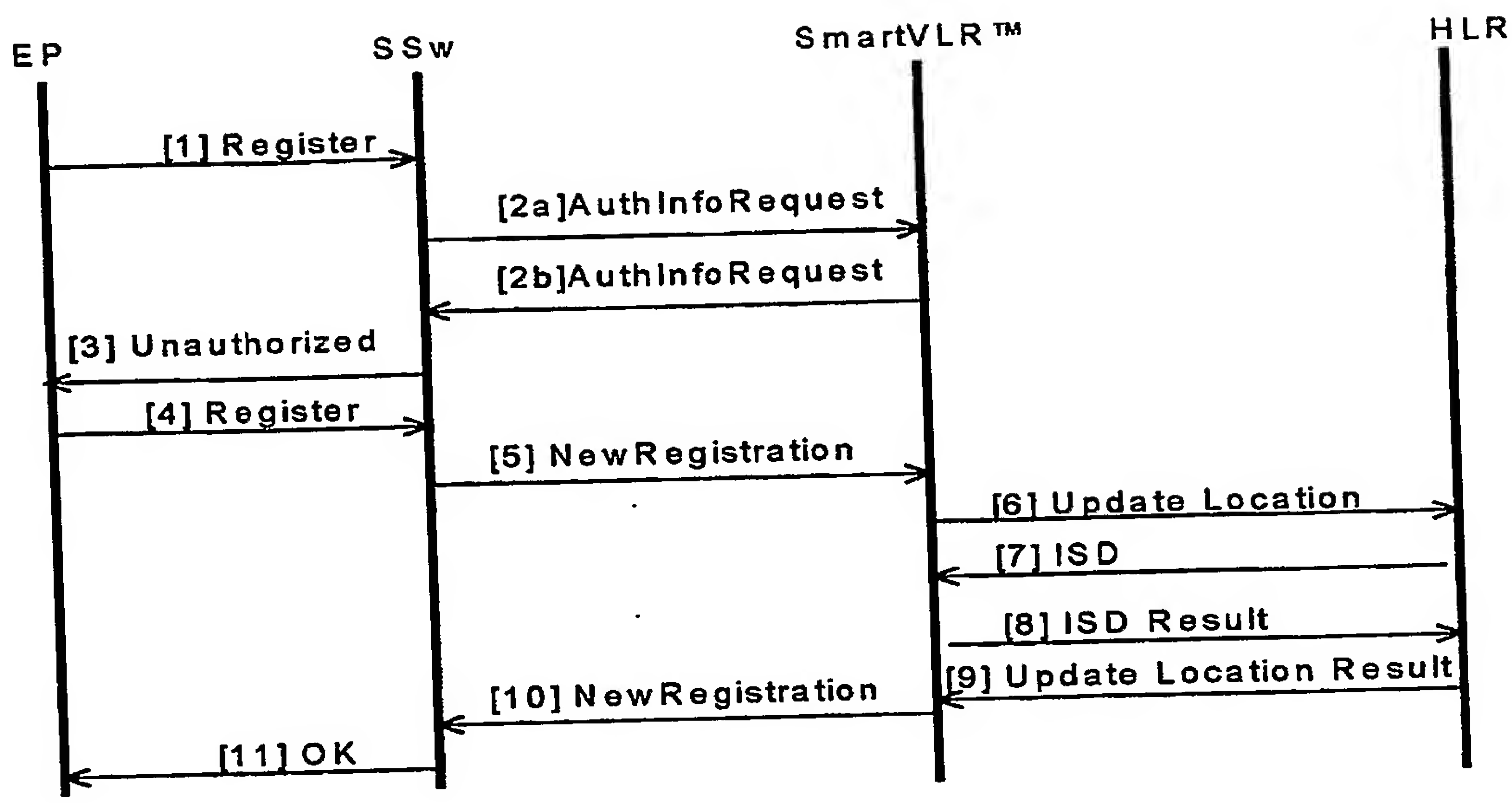
When a registration message arrived, the SSW makes an authentication by requesting the password for this endpoint from SmartVLR™ and comparing it with the password in the registration.

The SmartVLR™ stores the registered endpoint information including its IP Address. If a new registration message will arrive from a different address it will be rejected.

A registration message should be send periodically. This time is set by the endpoint in the registration message (the Expiration time). SSW can accept this period or it can force the endpoint reduce this value (send more recent registrations). The minimum value for expiration (registration period) is 10 seconds.
If no new registration message is arrived after the defined period – the endpoint is used as unregistered one.

- Every fixed (=VoIP) subscriber is registered in HLR just like any other GSM subscriber (including a unique IMSI and MSISDN)
- Fixed subscribers have TICK/OICK for the SmartSCP™ Fixed Mobile Convergence Service
- Fixed subscribers use SIP URI for SIP level signaling – sip: msisdn@orange.co.il
- Registration (emulates normal mobile attach):
  - User sends SIP REGISTER to SIP SERVER
  - SIP SERVER authenticates user password (in future, SIM based authentication may be employed)
  - SIP SERVER notifies the SmartVLR™
  - The SmartVLR™ sends LOCATION UPDATE message to HLR
  - The HLR sends INSERT SUBSCRIBER DATA message with subscriber profile back to the SmartVLR™

3.2.2 Diagram



EP
SSw
SmartVLR ™
HLR
[1] Register
[2a]AuthInfoRequest
[2b]AuthInfoRequest
[3] Unauthorized
[4] Register
[5] NewRegistration
[6] Update Location
[7] ISD
[8] ISD Result
[9] Update Location Result
[10] NewRegistration
[11] OK

3.2.3 Description:
This section describes the messages in the diagram, including sample messages for the main SIP messages.
For more details about the messages between the SSW unit and the SCP, see "FMC SSW and SCP interface" document.

1) <u>**Register**</u>
EP sends registration to SSW with the following information:
   - URI
   - Expiration time

   Sample message:
   REGISTER sip:192.168.40.62 SIP/2.0
   Via: SIP/2.0/UDP 194.90.115.62:5061
   From: <sip:0544160260@192.168.40.62:5061>
   To: <sip:0544160260@192.168.40.62:5061>
   Contact: "fixed260" <sip:0544160260@194.90.115.62:5061>
   Call-ID: 9FBEC6EC862D422D810FEAD4C336F7FE@192.168.40.62
   CSeq: 24054 REGISTER
   Expires: 120
   Content-Length: 0

2) <u>**AuthInforequest**</u>
   a. SSW sends request to SmartVLR™ including the following information:
      - IP Address
      - URI

   b. SmartVLR™ sends Authentication information. SSW authorizes message. In case no password or password is invalid or subscriber IP address is invalid SSW sends Reject to EP. The response includes the following information:
      - Password (if OK)
      - User Authentication Name (if OK)
      - Result (OK or error code)

3) <u>**Unauthorized**</u>
SSW reject the registration with error code = Authorization required.

   Sample message:
   SIP/2.0 401 Unauthorized
   From: <sip:0544160260@192.168.40.62:5061>
   To: <sip:0544160260@192.168.40.62:5061>;tag=3e28a8c0-13c4-401cef0f-1471b49d-5e1f
   Call-ID: 9FBEC6EC862D422D810FEAD4C336F7FE@192.168.40.62
   CSeq: 24054 REGISTER
   WWW-Authenticate: Digest realm="orange.co.il", nonce="9a7e9cd58266b68394", stale=true, algorithm=MD5
   Via: SIP/2.0/UDP 194.90.115.62:5061;received=192.168.40.62

4) <u>**Register**</u>

EP sends registration to SSW with Authorization including the following information:

- IP Address
- URI
- Expiration

Sample message:

```
REGISTER sip:192.168.40.62 SIP/2.0
Via: SIP/2.0/UDP 194.90.115.62:5061
From: <sip:0544160260@192.168.40.62:5061>
To: <sip:0544160260@192.168.40.62:5061>
Contact: "fixed260" <sip:0544160260@194.90.115.62:5061>
Call-ID: 9FBEC6EC862D422D810FEAD4C336F7FE@192.168.40.62
CSeq: 24055 REGISTER
Expires: 120
Authorization: Digest
username="User260",realm="orange.co.il",nonce="9a7e9cd58266b68394",response="04a6b7554d7f
b74f563e753e3ea02eb6",uri="sip:192.168.40.62",algorithm="MD5"
Content-Length: 0
```

5) **<u>New Registration</u>**

SSW sends request to check and store "Contact" headers and IP.
The request includes the following information:

- IP Address
- URI

6) **<u>Update Location</u>**

SmartVLR™ sends Update Location service toward the HLR.
UL sample Parameters:
Operation Code = Update Location
Update Location arguments Sequence:

- IMSI = 0300469832 (bcd)
- MSC Number = (International number, ISDN/telephony number plan) 97254120011
- VLR Number: (International number, ISDN/telephony number plan) 97254120011

7) **<u>ISD</u>**

HLR sends Insert Subscriber Data service toward SmartVLR™ to specify subscriber profile.
ISD sample Parameters:
Operation Code = Insert Subscriber Data
Invoke ID = 5
Insert Subscriber Data arguments Sequence:

- IMSI = 0300469832 (bcd)
    - MS ISDN - 97267773951
    - BearerServiceList:
        - GENERAL_DATACDA

- GENERAL_DATACDS
- Teleservice List:
    - TELEPHONY
    - SHORTMESSAGEMT_PP
    - SHORTMESSAGEMO_PP
    - FACSIMILEGROUP3ANDALTERSPEECH
    - AUTOMATICFACSIMILEGROUP3
- Provisioned SS List:
    - CALLBARRINGINFO: BOIC_EX_HC
    - SS_DATA: HOLD
    - SS_DATA: MULTIPTY
    - SS_DATA: CLIR
    - SS_DATA: FORWARDINGINFO_E
        - SS_CODE : CFB
        - TELESERVICE : ALLSPEECHSERVICES
        - FORWARD NUMBER: 97254151200
        - SUBADDRESS: 0x1F3E1280
- Extension container: 2A 86 3A 00 89 61 3A 01 00

8) <u>ISD Result</u>

SmartVLR™ sends Insert subscriber data response message toward HLR

ISD Result sample Parameters:

Invoke ID = 5

9) <u>Update Location result</u>

HLR sends response message toward SmartVLR™

Update Location result sample Parameters:

Invoke ID = 5

Operation Code = Update Location

Update Location result arguments Sequence:

- HLR NUMBER: 97254120031

10) <u>New Registration Result</u>

SmartVLR™ returns confirmation (or Reject), depends on Update Location Result, including the following information:

i. Result (OK or error code)

11) <u>OK</u>

SSW sends OK to EP (or Reject)

Sample message:

SIP/2.0 200 OK
From: <sip:0544160260@192.168.40.62:5061>
To: <sip:0544160260@192.168.40.62:5061>;tag=3e28a8c0-13c4-401cef0f-1471b597-5735
Call-ID: 9FBEC6EC862D422D810FEAD4C336F7FE@192.168.40.62

CSeq: 24055 REGISTER
Via: SIP/2.0/UDP 194.90.115.62:5061;received=192.168.40.62
Contact: "fixed260"<sip:0544160260@194.90.115.62:5061>;expires=120
Content-Length:0

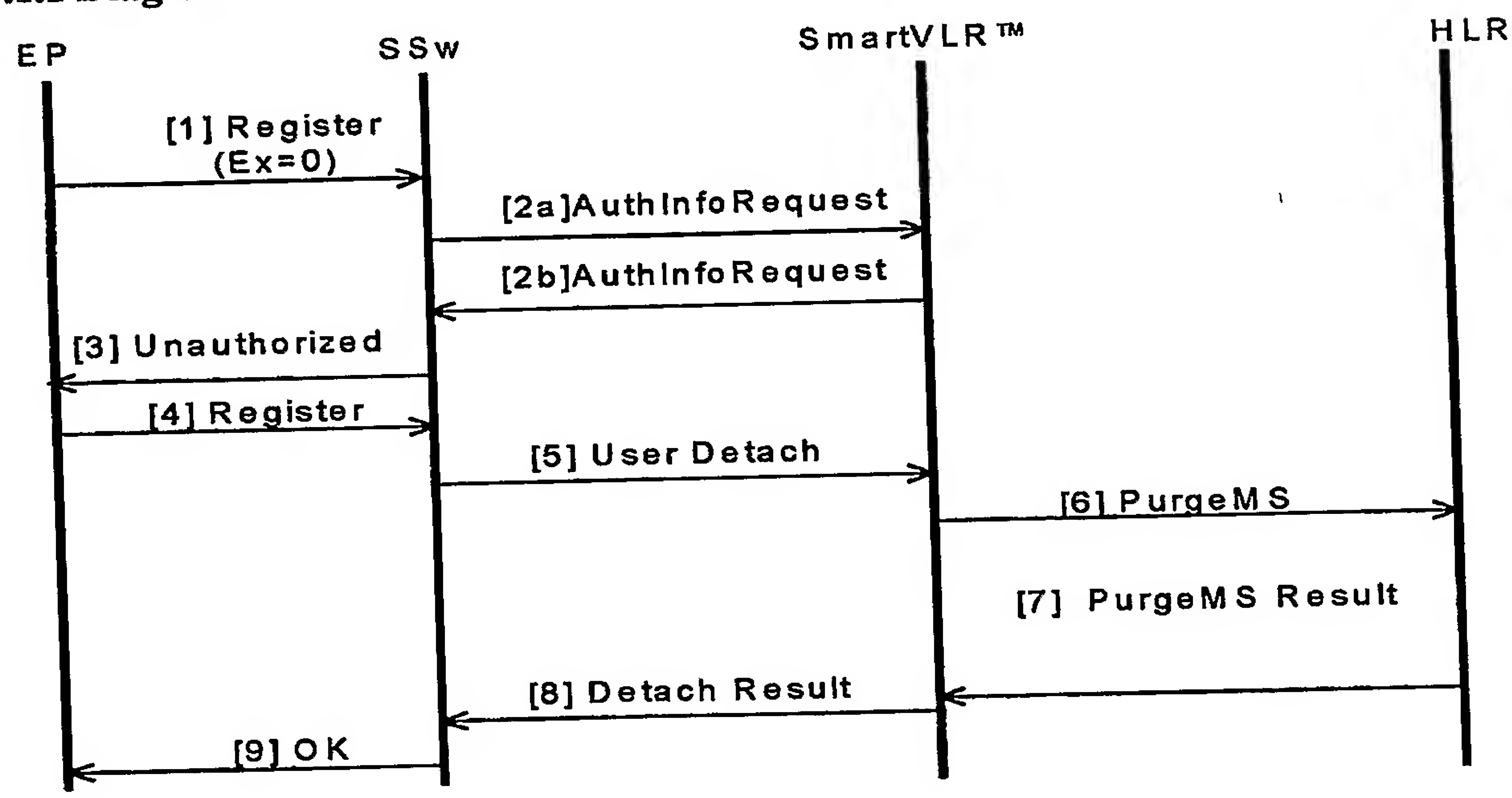
3.3 Deregistration

Basically, each endpoint registration has predefined period of validity. This time is determined during the registration procedure (the 'expire' parameter).
The end point may send an 'deregistration' message by sending a simple registration message with the parameter 'expire' set to 0.
The deregistration may initiated also by the HLR and the VLR.

3.3.1 Deregistration initiated by EP

This scenario describes Deregistration situation created by endpoint

3.3.1.1 Diagram



3.3.1.2 Description

This section describes the messages in the diagram, including sample messages for the main SIP messages.
For more details about the messages between the SSW unit and the SCP, see "FMC SSW and Smart SCP interface" document.

1) **Register**
EP sends registration to SSW with the following information:
- URI
- Expiration time=0

Sample message:
REGISTER sip:192.168.40.62 SIP/2.0
Via: SIP/2.0/UDP 194.90.115.62:5061
From: <sip:0544160260@192.168.40.62:5061>
To: <sip:0544160260@192.168.40.62:5061>
Contact: "fixed260" <sip:0544160260@194.90.115.62:5061>
Call-ID: 9FBEC6EC862D422D810FEAD4C336F7FE@192.168.40.62
CSeq: 24054 REGISTER
Expires: 0
Content-Length: 0

2) **AuthInforequest**
   a. SSW sends request to SmartVLR™ including the following information:
      - IP Address
      - URI

   b. SmartVLR™ sends Authentication information. SSW authorizes message. In case no password or password is invalid or subscriber IP address is invalid SSW sends Reject to EP. The response includes the following information:
      - Password (if OK)
      - User Authentication Name (if OK)
      - Result (OK or error code)

3) **Unauthorized**
SSW reject the registration with error code = Authorization required.

Sample message:
SIP/2.0 401 Unauthorized
From: <sip:0544160260@192.168.40.62:5061>
To: <sip:0544160260@192.168.40.62:5061>;tag=3e28a8c0-13c4-401cef0f-1471b49d-5e1f
Call-ID: 9FBEC6EC862D422D810FEAD4C336F7FE@192.168.40.62
CSeq: 24054 REGISTER
WWW-Authenticate: Digest realm="orange.co.il", nonce="9a7e9cd58266b68394", stale=true, algorithm=MD5
Via: SIP/2.0/UDP 194.90.115.62:5061;received=192.168.40.62
Content-Length:0

S M A R T

4) **Register**
EP sends registration to SSW with Authorization including the following information:

- URI
- Authentication credentials of a user agent.
- Expiration time

Sample message:
REGISTER sip:192.168.40.62 SIP/2.0
Via: SIP/2.0/UDP 194.90.115.62:5061
From: <sip:0544160260@192.168.40.62:5061>
To: <sip:0544160260@192.168.40.62:5061>
Contact: "fixed260" <sip:0544160260@194.90.115.62:5061>
Call-ID: 9FBEC6EC862D422D810FEAD4C336F7FE@192.168.40.62
CSeq: 24055 REGISTER
Expires: 0
Authorization: Digest
username="User260",realm="orange.co.il",nonce="9a7e9cd58266b68394",response="04a6b7554d7f b74f563e753e3ea02eb6",uri="sip:192.168.40.62",algorithm="MD5"
Content-Length: 0

5) **User Detach**
SSW sends request to the VLR to deregister the user. The message contains the user URI.

6) **PurgeMS**
PurgeMS sample Parameters:
Operation Code = PurgeMS
Invoke ID = 1
PurgeMS arguments Sequence:

- IMSI = 0300469832 (bcd)
- VLR Number: (International number, ISDN/telephony number plan) 97254120011

7) **PurgeMS result**
HLR sends response message toward SmartVLR™
ISD Result sample Parameters:
Invoke ID = 1

8) **User Detach Result**
SmartVLR™ returns confirmation (or Reject), depends on Update Location Result, including the following information:

- Result (OK or error code)

9) <u>OK</u>
SSW sends OK to EP (or Reject)

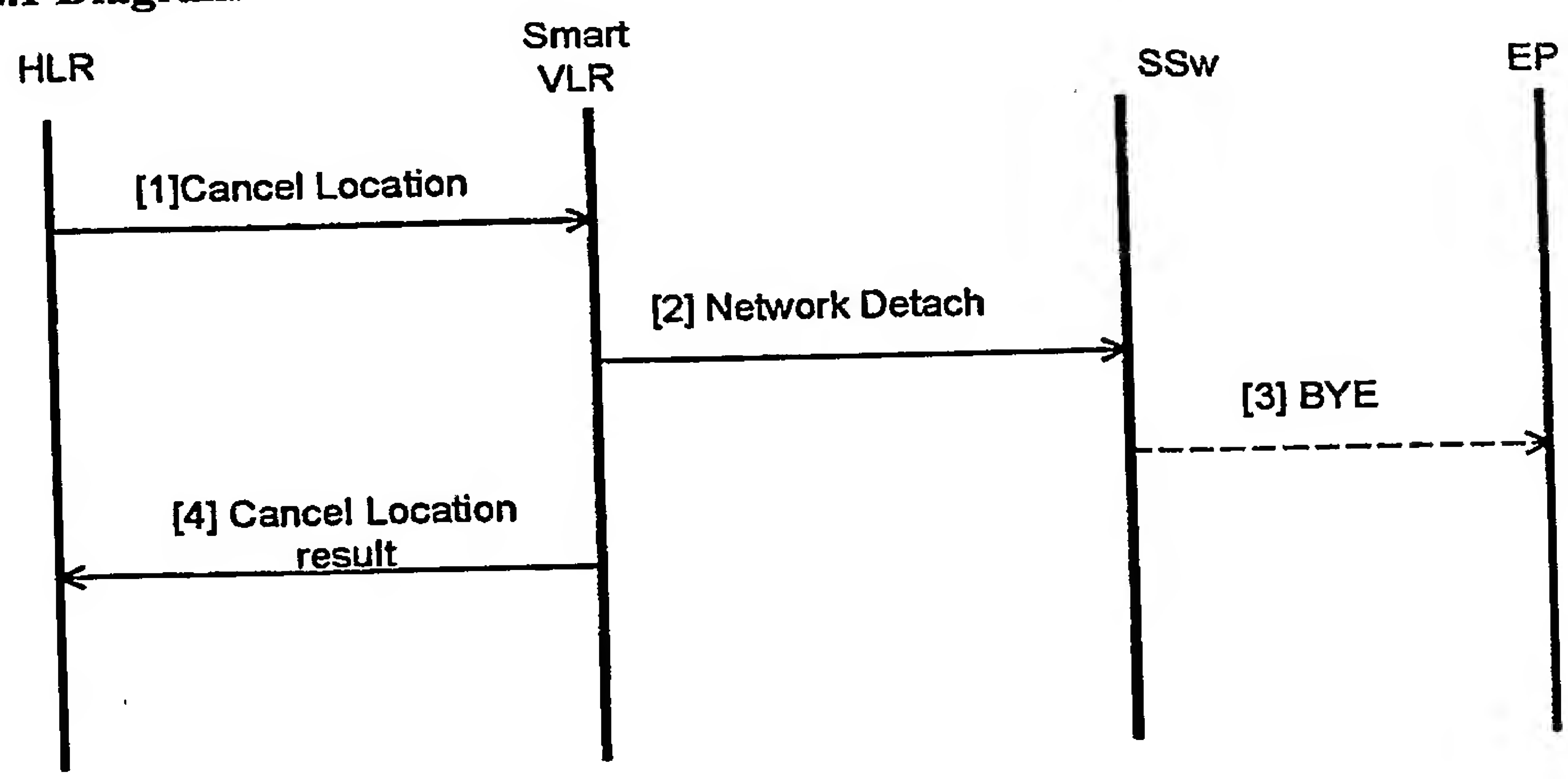
Sample message:

SIP/2.0 200 OK
From: <sip:0544160260@192.168.40.62:5061>
To: <sip:0544160260@192.168.40.62:5061>;tag=3e28a8c0-13c4-401cef0f-1471b597-5735
Call-ID: 9FBEC6EC862D422D810FEAD4C336F7FE@192.168.40.62
CSeq: 24055 REGISTER
Via: SIP/2.0/UDP 194.90.115.62:5061;received=192.168.40.62
Contact: "fixed260"<sip:0544160260@194.90.115.62:5061>;expires=120
Content-Length: 0

3.3.2 Deregistration initiated by HLR

This scenario describes Deregistration situation created by HLR, in case subscriber register to different VLR

3.3.2.1 Diagram



3.3.2.2 Description

This section describes the messages in the diagram, including sample messages for the main SIP messages.
For more details about the messages between the SSW unit and the SCP, see "FMC SSW and Smart SCP interface" document.

1) <u>Cancel Location (SS7)</u>
   Cancel Location sample Parameters:
   Operation Code = Cancel Location
   Invoke ID = 1
   Cancel Location arguments Sequence:
   - IMSI = 0300469832 (bcd)
   - MSISDN = 97254160272

2) <u>Network Detach</u>
   The VLR send detach message to the SSW with the user information.

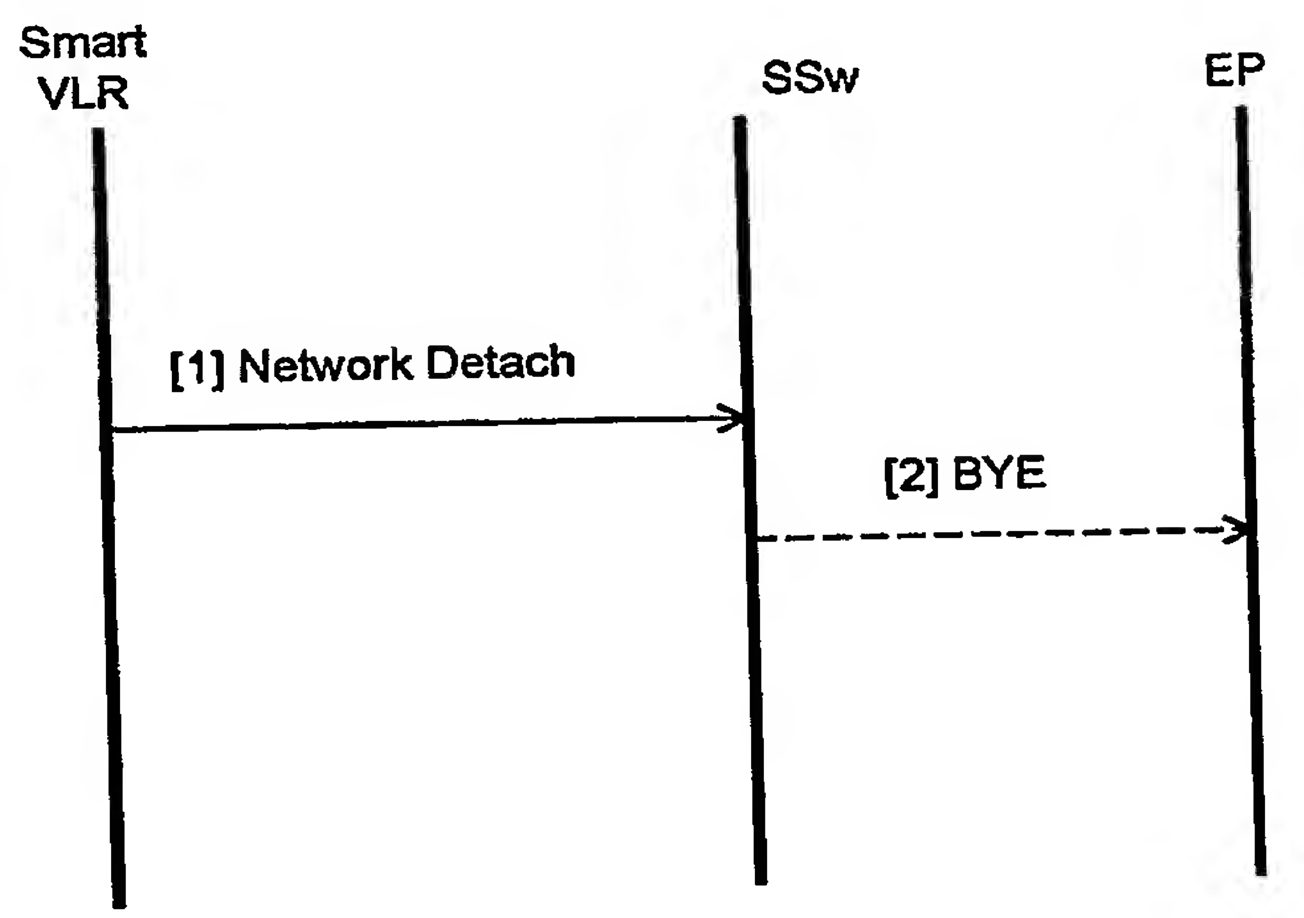
3) <u>BYE</u>
   If the user has an opens call the SSW sends BYE to the EP (close the call).

4) <u>Cancel Location result (SS7)</u>
   Cancel Location Result sample Parameters:
   Invoke ID = 1

3.3.3 Deregistration initiated by SmartVLR™

This scenario describes deregistration initiated by HLR.

3.3.3.1 Diagram



3.3.3.2 Description

This section describes the messages in the diagram, including sample messages for the main SIP messages.
For more details about the messages between the SSW unit and the SCP, see "FMC SSW and Smart SCP interface" document.

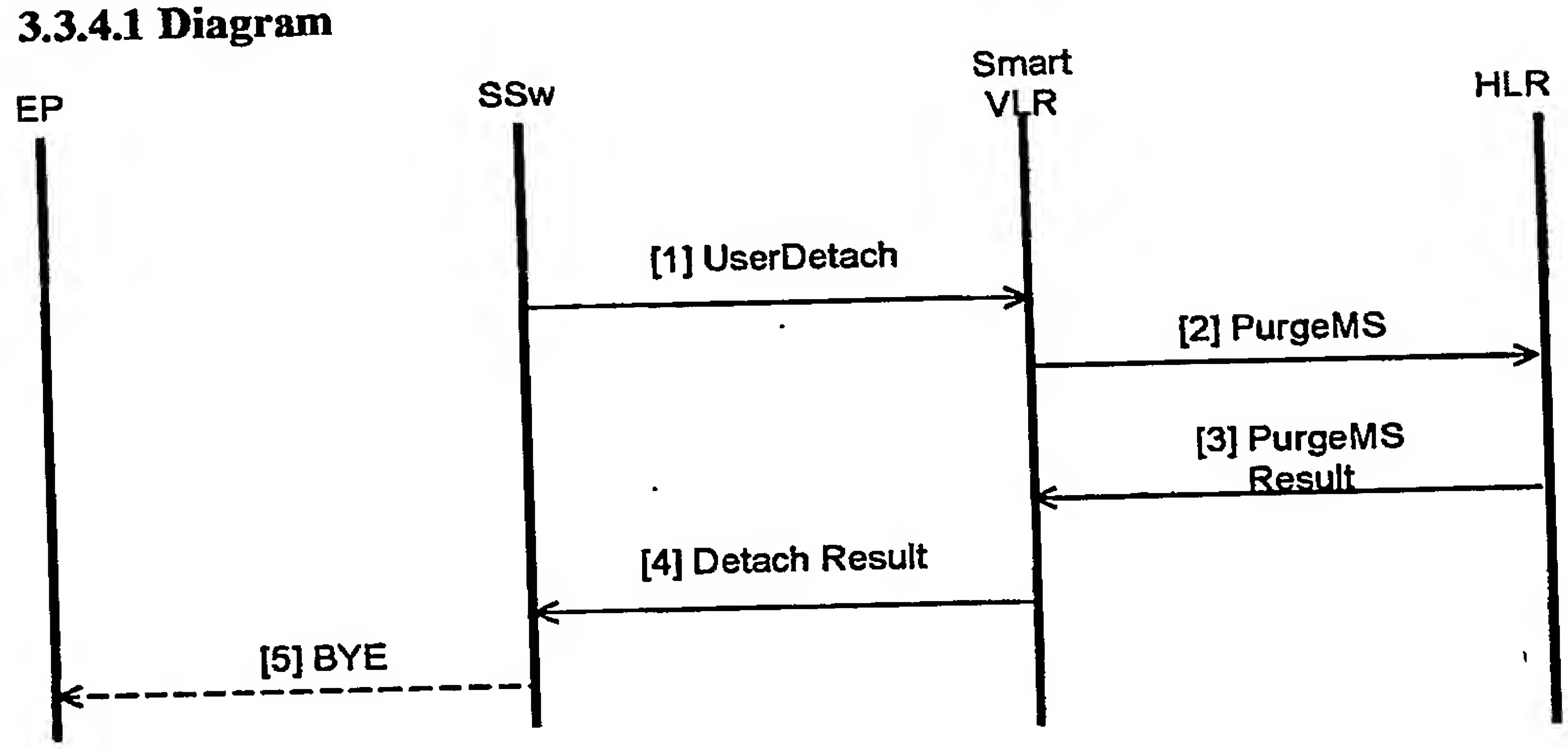
1) **<u>Network Detach</u>**
   The VLR send detach message to the SSW with the user information.

2) **<u>BYE</u>**
   If the user has an opens call the SSW sends BYE to the EP (close the call).

3.3.4 Deregistration initiated by SSW

This scenario describes Deregistration situation created by SSW

3.3.4.1 Diagram



3.3.4.2 Description

6) <u>**User Detach**</u>
SSW sends request to the VLR to deregister the user. The message contains the user URI.

7) <u>**PurgeMS**</u>
PurgeMS sample Parameters:
Operation Code = PurgeMS
Invoke ID = 1
PurgeMS arguments Sequence:
- IMSI = 0300469832 (bcd)
- VLR Number: (International number, ISDN/telephony number plan) 97254120011

8) <u>**PurgeMS Result**</u>
ISD Result sample Parameters:
Invoke ID = 1

9) <u>User Detach Result</u>
SmartVLR™ returns confirmation (or Reject), depends on Update Location Result, including the following information:

- Result (OK or error code)

10) <u>BYE</u>
If the user has an opens call the SSW sends BYE to the EP (close the call).

3.4 Call Initiation

3.4.1 Fix to mobile

3.4.1.1 General

Call may be initiated only by registered endpoint and only from the same IP address that was stored in the registration process.

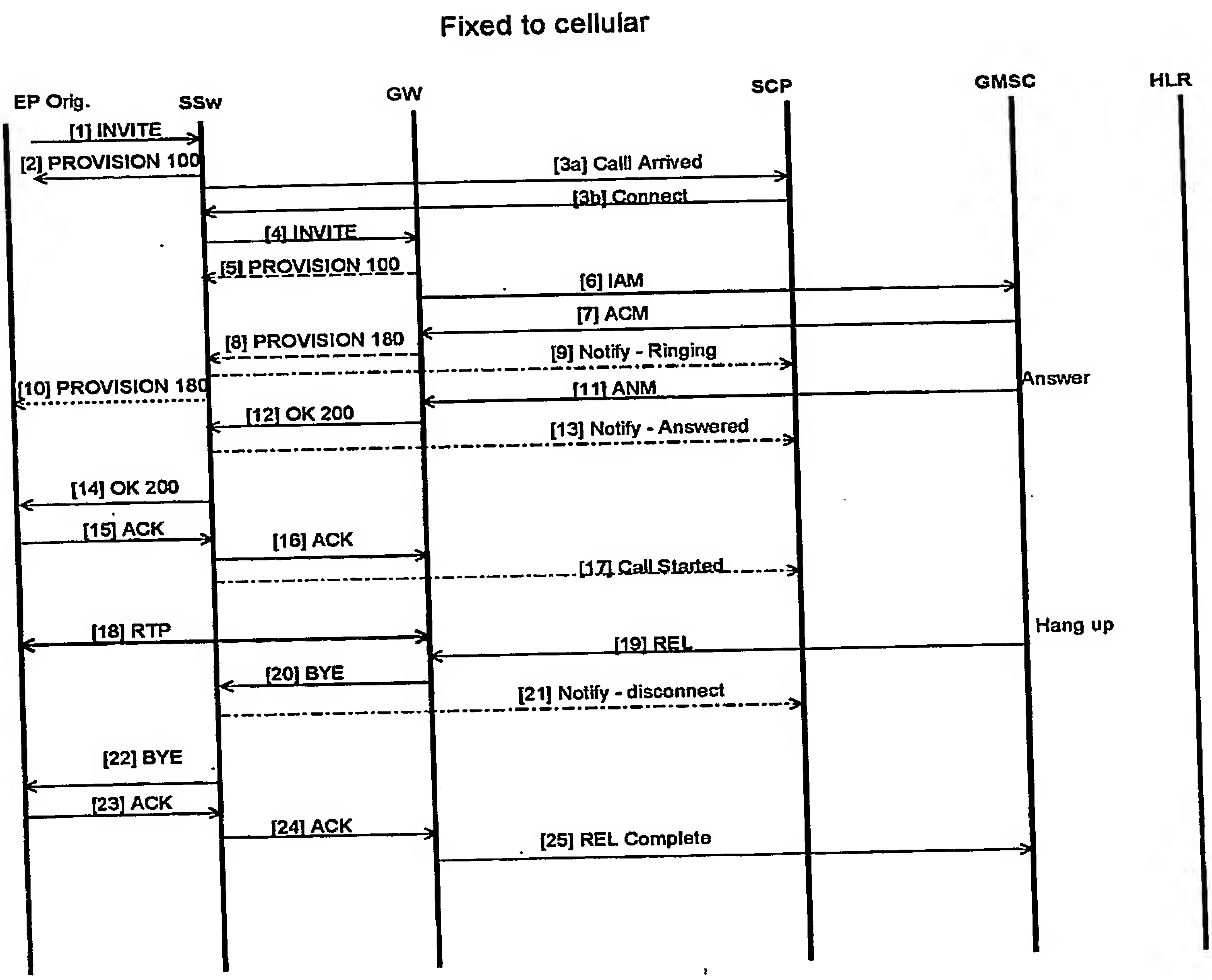
When an INVITE message arrived to the SSW it sends it to the SCP with the call information. The SCP is verifying that the endpoint is registered and that the caller IP identical to the IP address in the endpoint's stored registration.

The SCP sends to the SSW instructions for the arrival call. It may reject the call (if the caller is unregistered, if the caller IP address doesn't match the stored IP address, etc); it may supply new destination number and it may also change the originator information.

The SSW continues (or reject) the call according the instructions from the SCP.

In addition, the SSW will send to SCP information about the beginning and ending of the call; the SCP will forward this information to the HLR for billing.
CDR will be created also on the SSW.

3.4.1.2 Diagram



3.4.1.3 Description

This section supplies a detailed description of the messages in the diagram, including sample messages for the main SIP messages.
You can find more details, about the messages between the SSW unit and the SCP, in the "FMC SSW and Smart SCP interface" document.

1) <u>INVITE</u>
   EP sends INVITE to SSW including the following information:
   - To (MSISDN)

2) <u>PROVISION 100</u>
   SSW responses with PROVISION 100 message (acknowledge).

3) <u>Call Arrived</u>
   a. SSW sends request to SCP for caller and called information and passes the IP to verify that the called arrived from the register user.
      The message includes the following information:
      - To
      - From
      - IP Address of caller
      - Type of caller

   b. SCP checks the IP and the call information and return status and the requested information to the SSW. SCP recognize the 'To' field is not register in SmartVLR™, so in the returned 'Connect' message it indicate that the destination type is 'GSM', indicate the call should be deliver to GSM MSC through the GW. The response includes the following information:
      - To (can be new destination)
      - From (may be changed too)
      - Result (OK or error code)
      - Events – the events that SCP would like to be notified about.
      - Answer Timeout – how much time to wait in case that there is no answer.
      - Call properties – properties of the call (show/hide the originator phone number etc.).

4) **INVITE**
SSW sends INVITE to the GW including the following information:

- To (retrieved MSISDN)

Sample message:

```
INVITE sip:0544160272@192.168.40.10:5060 SIP/2.0
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-
19272449-30
Max-Forwards: 69
Supported: 0
Contact: <sip:0544160260@192.168.40.62:5062>
Content-Length:0
```

5) **PROVISION 100**
The GW reply with PROVISION 100 (acknowledge).
Sample message:

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-
19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Length: 0
```

6) **IAM**
The GW sends IAM to GMSC with CLI and DNIS

7) **ACM**
GMSC responses with ACM (acknowledge).

8) **PROVISION 180 [Optional]**
GW sends PROVISION 180 message (ringing), to SSW.

9) **Notify – Ringing [Optional]**
SSW sends to SCP RINGING notification (if it was requested to).

10) **PROVISION 180**

SSW passes the arrived PROVISION 180 message to the EP.

Sample message:

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Length: 0

11) **ANM**

When there is an answer – GMSC sends ANM to the GW.

12) **OK 200**

The GW sends OK 200 message to SSW.

Sample message:

SIP/2.0 200 Ok
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 297

13) **Notification – Answered [Optional]**

SSW sends to SCP ANSWER notification (if it was requested to).

14) **OK 200**

SSW sends OK 200 message to EP.

Sample message:

```
SIP/2.0 200 Ok
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-
19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 297
```

15) **ACK**

EP responses with ACK.

16) **ACK**

SSW passes the ACK to the GW.

17) **Call Started [Future]**

SSW send 'Call Started' message to SCP (for billing).

18) **RTP**

RTP Voice packets are streaming. Call is performed.

19) **REL**

REL command arrived from GMSC to GW.

20) **BYE**

GW stops the call and send BYE message to SSW.

21) **Call Closed [Optional]**

SSW sends to SCP DISCONNECT notification (if it was requested to).

SMART

22) <u>BYE</u>

SSW passes the BYE message to EP.

Sample message:

BYE sip:192.168.40.62;lr;rvRRParam=1921168340662 SIP/2.0
Via: SIP/2.0/UDP 192.168.40.10:5060
From: <sip:0544160272@192.168.40.62>;tag=116994780
To: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
Contact: <sip:0544160272@192.168.40.10:5060>
Route: <sip:0544160260@192.168.40.62:5062>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 43 BYE
Content-Length: 0

23) <u>ACK</u>

EP confirm with OK.

Sample message:

SIP/2.0 200 OK
From: <sip:0544160272@192.168.40.62>;tag=116994780
To: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 43 BYE
Via: SIP/2.0/UDP 192.168.40.10:5060;received=192.168.40.10
Supported: 0
Content-Length:0

24) <u>ACK</u>

SSW passes the OK to the GW.

Sample message:

SIP/2.0 200 OK
From: <sip:0544160272@192.168.40.62>;tag=116994780
To: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 43 BYE
Via: SIP/2.0/UDP 192.168.40.10:5060;received=192.168.40.10
Supported: 0
Content-Length:0

25) <u>REL Complete</u>

The GW sends Release complete message to GMSC.

3.4.2 Mobile to Fix

3.4.2.1 General

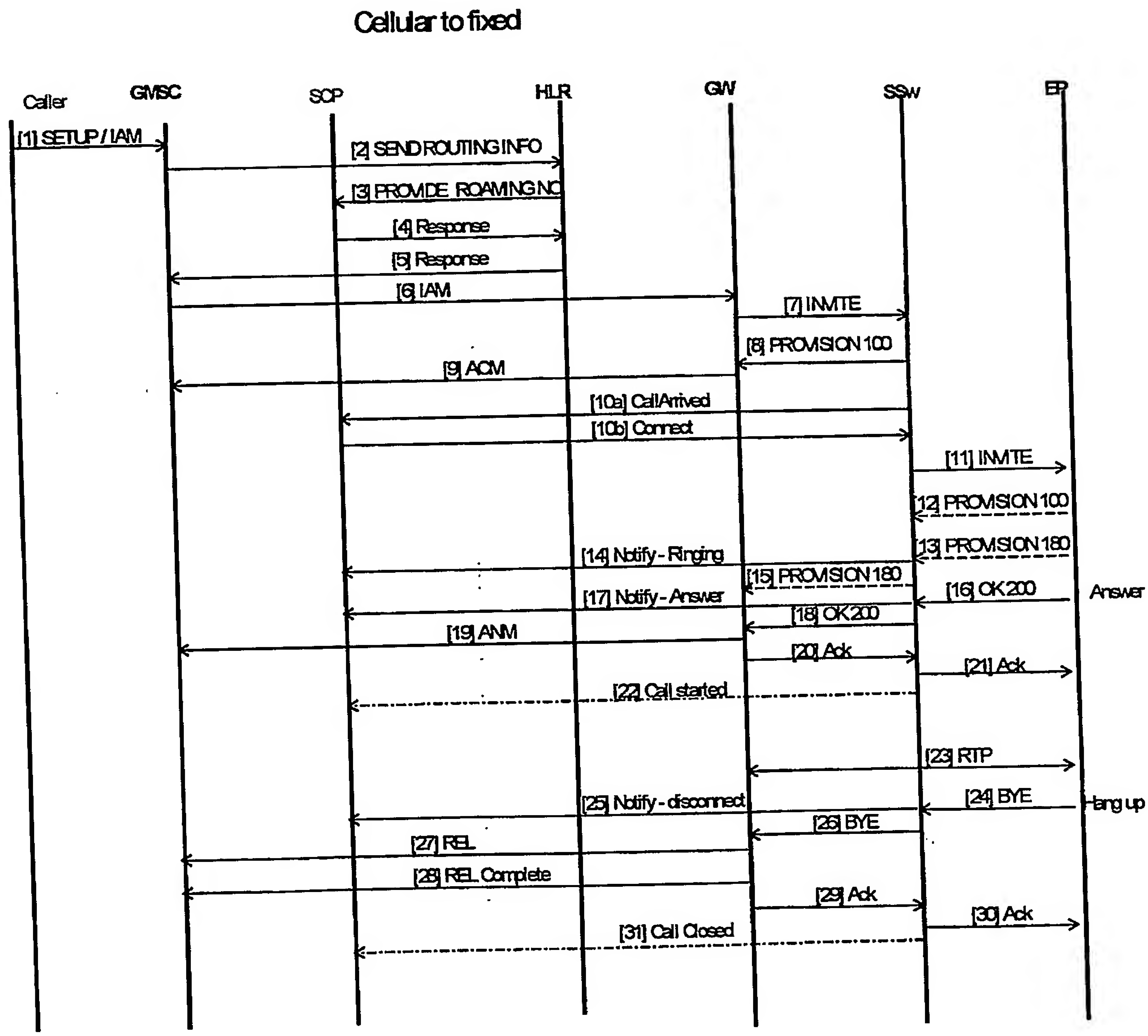
Normal Call Flow Concept

- GMSC asks the HLR for MSRN
- HLR asks the SmartVLR™ as a VLR for MSRN
- The SmartVLR™ return the GT for the MGW on the TSC as MSRN
- The HLR return the MSRN & TICK to GMSC
- GMSC triggers the SmartSCP™ using the TICK
- the SmartSCP™ invokes required terminating call control logic ("TICK") – multiple alerting, etc.
- If the final destination is a fixed subscriber – the call shall be routes using the MSRN towards the VoIP domain

Calls arrived from mobile to fixed can be established only if the endpoint is registered.
When an INVITE message received for endpoint, the SSW sends to the SCP the arrival MSRN and received the real MSISDN number for the arrival MSRN.
The SCP continues the call with the new MSISDN number.

In addition, the SSW will send to SCP information about the beginning and ending of the call; the SCP will forward this information to the HLR for billing.
CDR will be created also on the SSW.

3.4.2.2 Diagram



3.4.2.3 Description

This section supplies a detailed description of the messages in the diagram, including sample messages for the main SIP messages.
You can find more details, about the messages between the SSW unit and the SCP, in the "FMC SSW and Smart SCP interface" document.

1) **SETUP/IAM**
Caller send IAM message to GMSC with MSISDN number.

2) **SEND ROUTING INFO**
GMSC requests routing information for the retrieved MSISDN from the HLR.
SRI sample Parameters:
Operation Code = Send Routing Info
Invoke ID = 1
Send Routing Info arguments Sequence:
   - MS-ISDN
     - TBCD String : 994557890453
   - SM-RP-PRI: FALSE
   - ServiceCentreAddress
     - TBCD String : 905059910015

3) **PROVIDE ROAMING NO**
HLR recognize the MSISDN registered on SmartVLR™ , then requests roaming number from SmartVLR™ for provided IMSI.
PRN sample Parameters:
Operation Code = Provide roaming number
Invoke ID = 1
Provide roaming number arguments Sequence:
   - IMSI
     - TBCD String : 286015050494827
   - MSC-Number
     - TBCD String : 994550010000
   - LMSI: 00 05 7F 8D

4) **Response**
An SCP response with MSRN from predefined MSRN's to be routed to specific ISUP E1.
Provide roaming number Result sample Parameters:
Invoke ID = 1
Provide roaming number Result sequence parameter:
   - MSRN: 97254168000

5) <u>Response</u>
HLR send response to GMSC with the retrieved MSRN.
Send Routing Info Result sample Parameters:
Invoke ID = 1
Send Routing Info Result sequence parameter:

- MSISDN: 97254160219

6) <u>IAM</u>
GW received IAM from GMSC MSRN as Called party number.

7) <u>INVITE</u>
GW sends INVITE message to SSW.

Sample message:
INVITE sip:0544160272@192.168.40.10:5060 SIP/2.0
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
Max-Forwards: 69
Supported: 0
Contact: <sip:0544160260@192.168.40.62:5062>
Content-Length:0

8) <u>PROVISION 100</u>
SSW responses with PROVISION 100 message (acknowledge).

Sample message:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Length: 0

9) <u>ACM</u>
The GW sends ANM message to GMSC.

10) <u>Call Arrived</u>

a. SSW sends request to SCP for caller and called information and passes the IP to verify that the called arrived from the register user.
   The message includes the following information:
   - To
   - From
   - IP Address of caller
   - Type of caller

b. SCP checks call information and return status and the requested information to the SSW. SCP recognize the 'To' field is registered in SmartVLR™, so in the returned 'Connect' message it indicate that the destination type is 'Fixed'. The response includes the following information:
   - To MSISDN (corresponds with the arrival MSRN – if exists)
   - From (may be changed too)
   - Result (OK or error code)
   - Events – the events that SCP would like to be notified about.
   - Answer Timeout – how much time to wait in case that there is no answer.
   - Call properties – properties of the call (show/hide the originator phone number etc.).

11) <u>INVITE</u>

SSW send INVITE message (with the MSISDN) to the EP.

Sample message:

```
INVITE sip:0544160272@192.168.40.10:5060 SIP/2.0
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-
19272449-30
Max-Forwards: 69
Supported: 0
Contact: <sip:0544160260@192.168.40.62:5062>
Content-Length:0
```

12) **PROVISION 100**

EP responses with PROVISION 100 (Acknowledge).

Sample message:

SIP/2.0 100 Trying
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Length: 0

13) **PROVISION 180 [optional]**

EP sends status PROVISION 180 (ringing) to SSW.

Sample message:

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Length: 0

14) **Notify – Ringing [Optional]**

SSW sends to SCP RINGING notification (if it was requested to).

15) **PROVISION 180**

SSW passes the message to the GW.

Sample message:

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Length: 0

16) **OK 200**

S M A R T

User answered. EP sends OK 200 to SSW.

Sample message:
SIP/2.0 200 Ok
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 297

17) **Notify – Answer [Optional]**
SSW sends to SCP ANSWER notification (if it was requested to).

18) **OK 200**
SSW passes the OK 200 to the GW.

Sample message:
SIP/2.0 200 Ok
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 297

19) **ANM**
The GW sends ANM message to GMSC.

20) **ACK**
The GW confirm with ACK.

21) **ACK**
SSW passes the ACK to the EP.

22) **Call Started [Future]**
SSW send 'Call Started' message to SCP (for billing).

23) **RTP**
Call is performed.

24) **BYE**

User Hang up. BYE message is sent from the EP to SSW.

Sample message:
BYE sip:192.168.40.62;lr;rvRRParam=1921168340662 SIP/2.0
Via: SIP/2.0/UDP 192.168.40.10:5060
From: <sip:0544160272@192.168.40.62>;tag=116994780
To: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
Contact: <sip:0544160272@192.168.40.10:5060>
Route: <sip:0544160260@192.168.40.62:5062>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 43 BYE
Content-Length: 0

25) <u>Notify – Disconnect [Optional]</u>
SSW sends to SCP DISCONNECT notification (if it was requested to).

26) <u>BYE</u>
SSW passes the BYE message to the GW.

Sample message:
BYE sip:192.168.40.62;lr;rvRRParam=1921168340662 SIP/2.0
Via: SIP/2.0/UDP 192.168.40.10:5060
From: <sip:0544160272@192.168.40.62>;tag=116994780
To: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
Contact: <sip:0544160272@192.168.40.10:5060>
Route: <sip:0544160260@192.168.40.62:5062>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 43 BYE
Content-Length: 0

27) <u>REL</u>
The GW sends release message to GMSC.

28) <u>REL Complete</u>
GMSC confirm with release complete message.

29) <u>ACK</u>
GW confirms with ACK.

30) <u>ACK</u>

SSW passes the ACK to the EP.

Sample message:

SIP/2.0 200 OK
From: <sip:0544160272@192.168.40.62>;tag=116994780
To: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 43 BYE
Via: SIP/2.0/UDP 192.168.40.10:5060;received=192.168.40.10
Supported: 0
Content-Length: 0

31) **<u>Call Closed [Future]</u>**

SSW send 'Call Closed' message to SCP (for billing).

3.4.3 Fix to fix

3.4.3.1 General

Call may be initiated only by registered endpoint and only from the same IP address that was stored in the registration process.

Calls arrived from can be established only if the destination endpoint is registered too.

When an INVITE message arrived to the SSW it sends it to the SCP with the call information. The SCP is verifying that the endpoint is registered and that the caller IP identical to the IP address in the endpoint's stored registration.

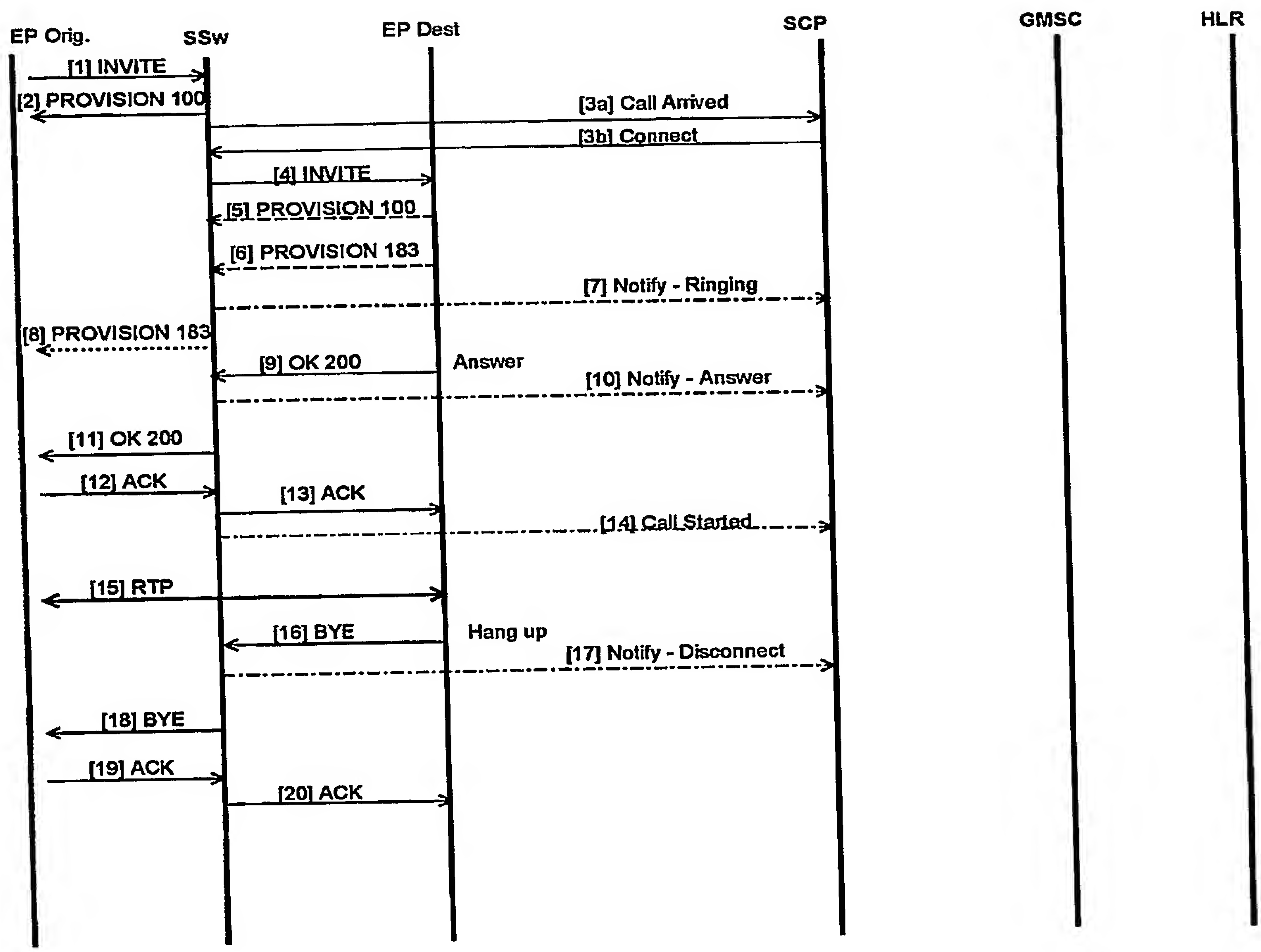
The SCP sends to the SSW instructions for the arrival call. It may reject the call (if the caller is unregistered, if the caller IP address doesn't match the stored IP address, etc); it may supply new destination number and it may also change the originator information.

The SSW continues (or reject) the call according the instructions from the SCP.

In addition, the SSW will send to SCP information about the beginning and ending of the call; the SCP will forward this information to the HLR for billing.
CDR will be created also on the SSW.

3.4.3.2 Diagram

Fixed to Fixed



3.4.3.3 Description

This section supplies a detailed description of the messages in the diagram, including sample messages for the main SIP messages.
You can find more details, about the messages between the SSW unit and the SCP, in the "FMC SSW and Smart SCP interface" document.

1) **INVITE**
EP1 sends INVITE to SSW including the following information:

- To (MSISDN)

Sample message:
INVITE sip:0544160272@192.168.40.10:5060 SIP/2.0
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
Max-Forwards: 69
Supported: 0
Contact: <sip:0544160260@192.168.40.62:5062>
Content-Length:0

2) **<u>PROVISION 100</u>**
SSW responses with PROVISION 100 message (acknowledge).

Sample message:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Length: 0

3) **<u>Call Arrived</u>**
   a. SSW sends request to SCP for caller and called information and passes the IP to verify that the called arrived from the register user.
   The message includes the following information:
      - To
      - From
      - IP Address of caller
      - Type of caller

b. SCP checks call information and return status and the requested information to the SSW. SCP recognize the 'To' field is registered in SmartVLR™, so in the returned 'Connect' message it indicate that the destination type is 'Fixed'. The response includes the following information:
   - To MSISDN (corresponds with the arrival MSRN – if exists)
   - From (may be changed too)
   - Result (OK or error code)
   - Events – the events that SCP would like to be notified about.
   - Answer Timeout – how much time to wait in case that there is no answer.
   - Call properties – properties of the call (show/hide the originator phone number etc.).

4) **INVITE**
SSW sends INVITE to the EP2 including the following information:
- To

Sample message:
INVITE sip:0544160272@192.168.40.10:5060 SIP/2.0
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Via: SIP/2.0/UDP 192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
Max-Forwards: 69
Supported: 0
Contact: <sip:0544160260@192.168.40.62:5062>
Content-Length:0

5) **PROVISION 100**
EP2 replies with PROVISION 100 (acknowledge).

Sample message:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.40.62:5060;branch=z9hG4bKKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Length: 0

6) **PROVISION 180 [Optional]**

EP2 sends PROVISION 180 message (ringing), to SSW.

Sample message:
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Length: 0

7) **<u>Notify – Ringing [Optional]</u>**
SSW sends to SCP RINGING notification (if it was requested to).

8) **<u>PROVISION 180</u>**
SSW passes the arrived PROVISION 180 message to the EP1.

Sample message:
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Length: 0

9) **<u>OK 200</u>**
EP2 sends OK 200 message to SSW (when there is an answer).

Sample message:
SIP/2.0 200 Ok
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 297

10) **<u>Notify – Answer [Optional]</u>**
SSW sends to SCP ANSWER notification (if it was requested to).

11) <u>OK 200</u>

SSW sends OK 200 message to EP1.

Sample message:

SIP/2.0 200 Ok
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGyGuiCUMu8U0m2W2WWC!kG0KY.1-190aa5c
Via: SIP/2.0/UDP 192.168.40.62:5062;received=192.168.40.62;branch=z9hG4bK-401e23a3-19272449-30
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Contact: <sip:0544160272@192.168.40.10:5060>
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 297

12) <u>ACK</u>

EP1 responses with ACK.

Sample message:

ACK sip:0544160272@192.168.40.10:5060 SIP/2.0
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 ACK
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGEOuiCUMu8U0m2W2WWC!kG0KY.1
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Via: SIP/2.0/UDP 192.168.40.62:5062;branch=z9hG4bK-401e23a9-19273dad-64a9
Max-Forwards: 69
Contact: <sip:0544160260@192.168.40.62:5062>
Content-Type: application/SDP
Content-Length:297

SMART

13) <u>ACK</u>

SSW passes the ACK to the EP2.

Sample message:

ACK sip:0544160272@192.168.40.10:5060 SIP/2.0
From: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
To: <sip:0544160272@192.168.40.62>;tag=116994780
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 1 ACK
Via: SIP/2.0/UDP
192.168.40.62:5060;branch=z9hG4bKkGyGkGyGkGEOuiCUMu8U0m2W2WWC!kG0KY.1
Record-Route: <sip:192.168.40.62;lr;rvRRParam=1921168340662>
Via: SIP/2.0/UDP 192.168.40.62:5062;branch=z9hG4bK-401e23a9-19273dad-64a9
Max-Forwards: 69
Contact: <sip:0544160260@192.168.40.62:5062>
Content-Type: application/SDP
Content-Length: 297

14) <u>Call Started [Future]</u>

SSW send 'Call Started' message to SCP (for billing).

15) <u>RTP</u>

Call is performed.

16) <u>BYE</u>

EP2 stops the call and send BYE message to SSW.

Sample message:

BYE sip:192.168.40.62;lr;rvRRParam=1921168340662 SIP/2.0
Via: SIP/2.0/UDP 192.168.40.10:5060
From: <sip:0544160272@192.168.40.62>;tag=116994780
To: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
Contact: <sip:0544160272@192.168.40.10:5060>
Route: <sip:0544160260@192.168.40.62:5062>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 43 BYE
Content-Length: 0

17) <u>Notify – disconnect [Optional]</u>

SSW sends to SCP DISCONNECT notification (if it was requested to).

18) <u>BYE</u>

SSW passes the BYE message to EP1.

Sample message:

BYE sip:192.168.40.62;lr;rvRRParam=1921168340662 SIP/2.0
Via: SIP/2.0/UDP 192.168.40.10:5060
From: <sip:0544160272@192.168.40.62>;tag=116994780
To: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
Contact: <sip:0544160272@192.168.40.10:5060>
Route: <sip:0544160260@192.168.40.62:5062>
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 43 BYE
Content-Length: 0

19) <u>ACK</u>

EP1 confirm with OK.

Sample message:

SIP/2.0 200 OK
From: <sip:0544160272@192.168.40.62>;tag=116994780
To: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 43 BYE
Via: SIP/2.0/UDP 192.168.40.10:5060;received=192.168.40.10
Supported: 0
Content-Length:0

20) <u>ACK</u>

SSW passes the OK to EP2.

Sample message:

SIP/2.0 200 OK
From: <sip:0544160272@192.168.40.62>;tag=116994780
To: <sip:0544160260@192.168.40.62:5062>;tag=0-13c6-401e23a3-19272449-4973
Call-ID: a4d9ac-0-13c6-401e23a3-19272449-31e2@192.168.40.62
CSeq: 43 BYE
Via: SIP/2.0/UDP 192.168.40.10:5060;received=192.168.40.10
Supported: 0
Content-Length:0

3.5 HLR VLR Interface

SmartVLR™ will wait for following operation received from HLR:

1) Provide Roaming Number (in case of incoming call fro mobile to fix) HLR will send this operation to SmartVLR™. If subscriber is register then SmartVLR™ will send response TCAP Return Result message with MSRN parameter.
   If the subscriber is not register SmartVLR™ will response TCAP Return Error.
2) Insert Subscriber data.
   In case that profile of subscriber has been added in HLR, the HLR will invoke this operation to SmartVLR™.
   If the subscriber is registrate SmartVLR™ will response TCAP Return Result.
   Note - This operation is invoked after SmartVLR™ sends update location operation
3) Delete subscriber data
   In case that profile of subscriber has been removed in HLR, the HLR will invoke this operation to SmartVLR™.
   If the subscriber is registrate SmartVLR™ will response TCAP Return Result.
4) Cancel location
   In case Sip subscriber is register in another VLR , or partner network decides to remove this subscriber , this operation will invoke to SmartVLR™.
   SmartVLR™ will delete this sibscriber profile and notice SSW by Network Detach message
5) HLR_RESET
   This message is invoked by HLR after reset of HLR. When SmartVLR™ receives this operation it will derive all involved MSs of that HLR either from the HLR Identity List (if present), or from the HLR number. The SmartVLR™ will then mark these MSs with the indicator "Location Information Confirmed in HLR" set to "Not Confirmed" and will deactivate all subscriber tracings for these Mss. Then SmartVLR™ will invoke Update location service for each of those MSs.

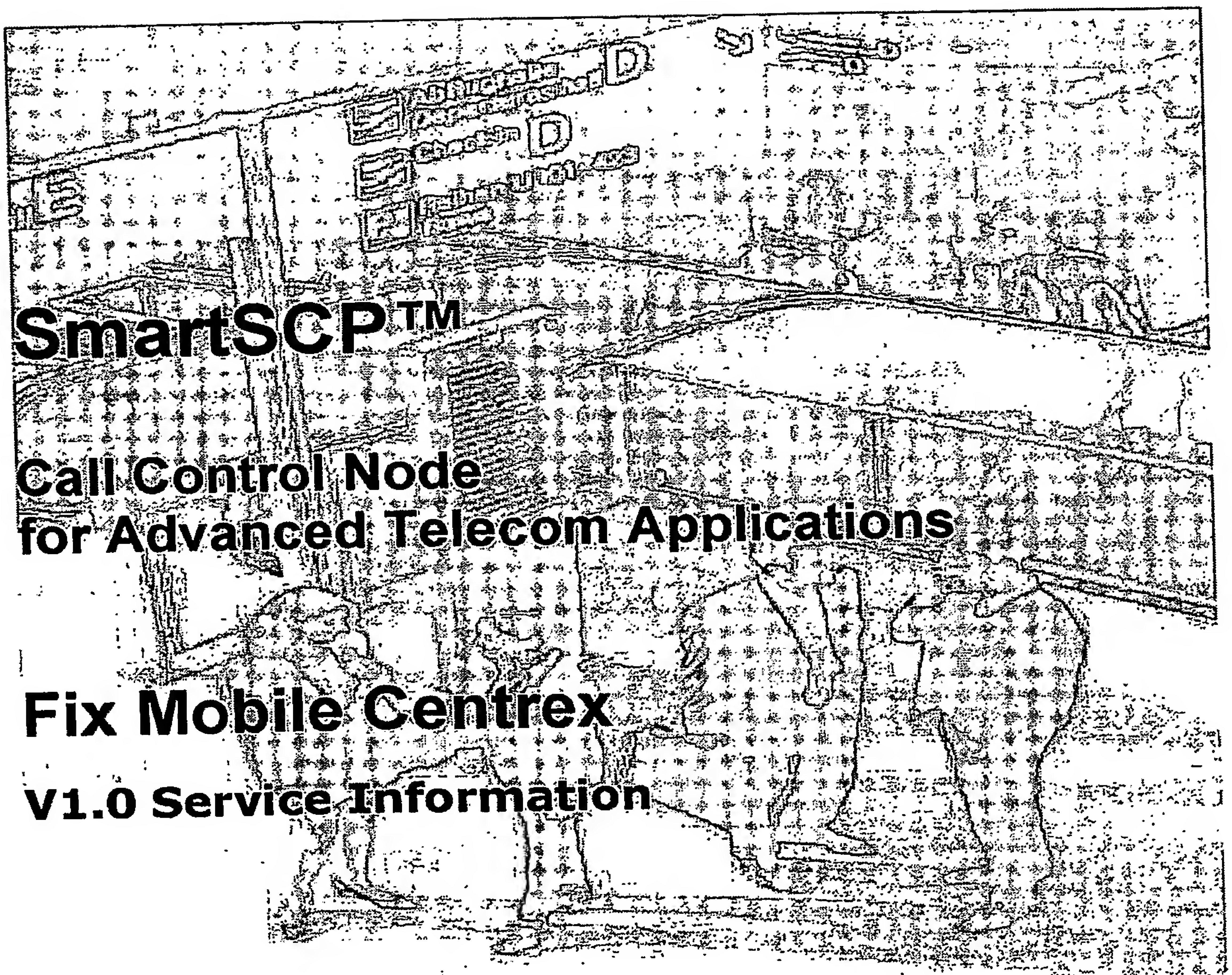
Abstract

Enables the operator to offer a complete Centrex solution including long distance services for a business, where all phones – fixed and mobile – enjoy the same features such as private numbering plans.

The solution is fully hosted by the mobile operator on its premises, and integrates with the business' premises infrastructure, existing (PBX) or new (VoIP). The solution is composed of known components – soft-switch, VoIP media gateways and the SmartSCP™ - deployed in the mobile operator's premises, interconnected and enhanced to achieve required functionality.

In the business premises, standards VoIP equipment is used, whether to enable VoIP for legacy PBX telephony or to support new native IP telephony components. The Business is connected to the mobile operator's core network by a common broadband IP, or even voice-enabled-only Fixed Wireless Terminals ("tellular").

SMART



SmartSCP™

Call Control Node for Advanced Telecom Applications

Fix Mobile Centrex

V1.0 Service Information

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Service Description

A converged fixed mobile Centrex solution (FMC) offers two main propositions: ease-of-use as a higher productivity factor for end-users, and rendering fixed services as well as mobile services to get a larger business expenditure share by mobile operators.

Centrex is a suite of services provided to a business by a telecommunications service provider, as a replacement of a business owned, self-operated and self-managed private telecommunications infrastructure (e.g., Private Branch Exchange PBX). Centrex infrastructure is installed by the service providers, at their premises, while users use their terminals (phones) normally, as though connected to a local PBX. Centrex mitigates the CAPEX (Capital Expenditures) associated with procurement and deployment of internal infrastructure, from the business to the service provider, as well as the OPEX (Operational Expenditures) associated with internal telecom management. Mobile Centrex is a similar suite of services (such as private numbering plans or closed user groups) applied to the business' mobile phones (sometimes referred to as MVPN – Mobile Virtual Private Network).

While both Centrex and Mobile Centrex are successful services, each is operated by a different type of service provider – Centrex by ILECs (Incumbent Local Exchange Carriers) and CLECs (Competitive Local Exchange Carriers), and Mobile Centrex by MNOs (Mobile Network Operators) and MVNOs (Mobile Virtual Network Operators). As business users use both fixed and mobile phones, a unification of Centrex service is called for.

The solution will enable mobile operators to offer their existing business customers (as well as prospect customers) a simple, unified but outsourced management of their organizational network. Businesses will enjoy Centrex benefits regardless of existing telecom configuration (with or without PBX, IP based or traditional). Businesses will be able to gradually migrate from a PBX-based telephony environment to an IP Telephony environment, with enhanced features and reduced CAPEX and OPEX.

The solution integrates voice-over-IP (VoIP) soft-switch (the call control functional component in VoIP networks), with a service control point (SCP, the call control functional component in SS7 based telecom networks, such as GSM or cdma2000). This innovative solution, seamless and transparent to the user, is unique in coordinating two different technologies and paradigms – SS7 and SIP.

Functional Features:

- Full and seamless integration between Mobile Centrex and SIP-based IP Centrex
  - Private numbering plans
  - Closed user groups
  - Hierarchical organizational structures
  - Black and white lists for organizational groups and individuals
  - Multiple incoming call routing profiles per user
  - Multiple accounts and identifiers per user
  - Call records
  - Simplified management
- Delivering Converged Centrex to multiple organizations, sizing from SOHO, through SME up to corporate.
- Support of different mobile networks: GSM, cdma2000, UMTS
- Support of different transport mechanisms between the business and the service provider: landline, fixed wireless terminals (FWT) voice, FWT DATA (for UMTS and cdma2000)
- Support of multiple terminals: mobile phones, fixed "POTS" phones, fixed smart phones, IP phones, computer IP Telephony clients, PBX
- Support of direct connectivity from the business to other operators

Key Benefits:

To mobile operators (MNOs and MVNOs):

- Differentiation from competitors through innovative services
- Higher customer attraction due to the higher perceived value of the solution
- Higher customer ARPU due to the higher portion of communications handled by the operator
- Higher customer loyalty and lower churn due to the large portion of customer management data stored in the mobile operator's infrastructure
- Indirect yet efficient and aggressive entry to the fixed telecom business for higher revenues

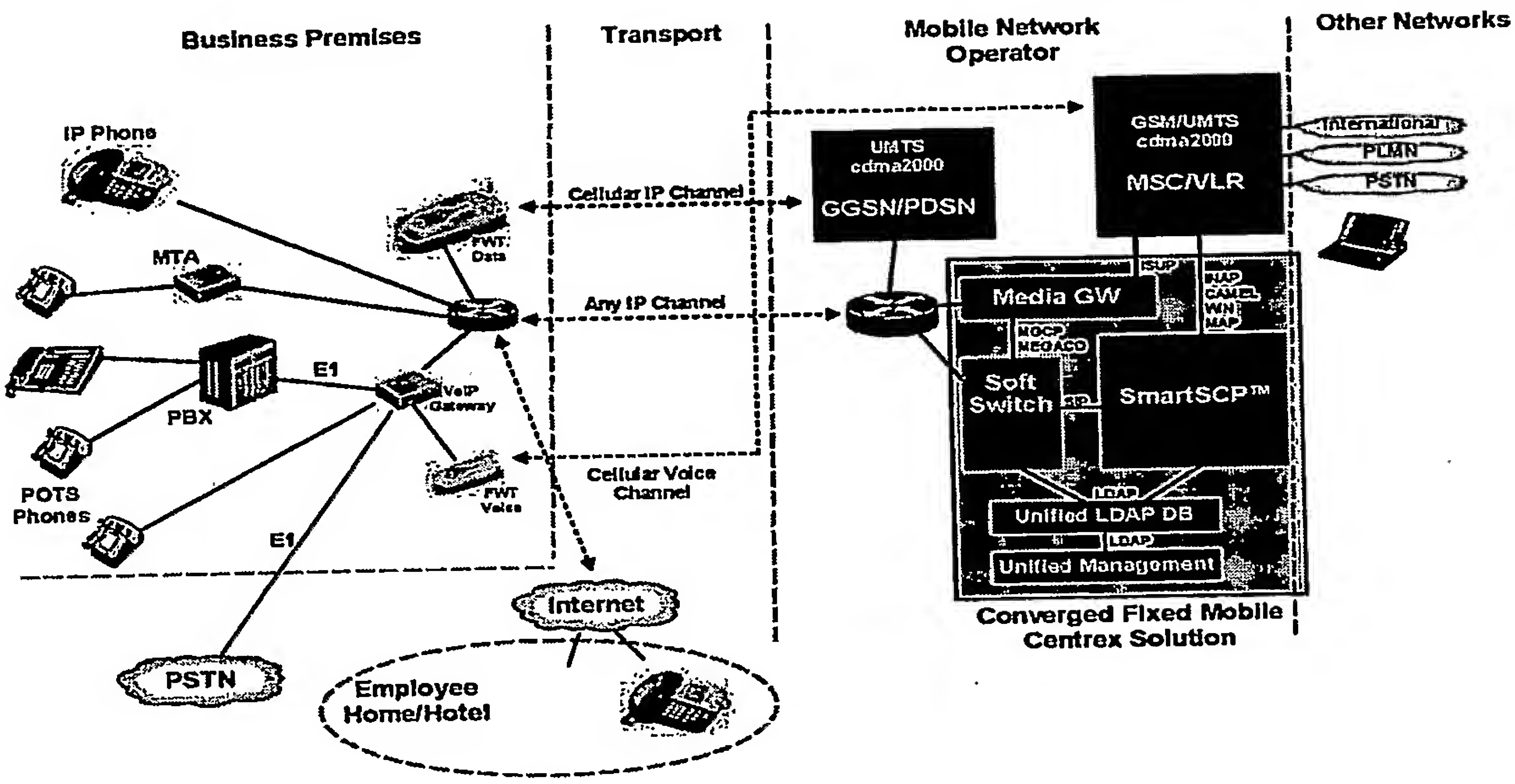
For the End-User (Centrex in General)

- No investments (CAPEX and OPEX) in non-core activities – service provider owns, installs, configures, operates, maintains and continuously upgrades the infrastructure in its own premises
- Very high availability and reliability as infrastructure is Telco-grade and operated 24X7 by service provider
- Simplicity - as service provider configures the service, and customer is provided with easy to use self-management interfaces
- Scalability as businesses can purchase exactly the number of "lines" needed, at the time they need them
- Extensive feature portfolio to the smallest of businesses, as to the largest

To End-Users (for Converged Centrex):

- Anytime and anywhere availability of all phone numbers
- Same features (own number, numbering plans, etc.) in multiple terminals –choose the best suited terminal at any given moment
- Single voice mailbox
- Simple, easy to use self management by Web or Mobile Phone
- One bill from one provider
- One customer service number to call in case of trouble

Product Architecture:



The diagram shows the overall solution's architecture.

- At the mobile operator's premises, the following platforms are depicted:

1. MSC/VLR – the existing voice switch
2. GGSN/PDSN – the mobile data IP switch (when using wireless broadband IP connectivity as VoIP bearer between the business and the operator)

- In addition, the following platforms are integrated for deploying the solution:

1. The SmartSCP™ - a service control point where all service logic is executed to control entire aspects of call routing and control, for both fixed and mobile terminals.
2. Soft switch – providing a call control interface intermediary towards the SmartSCP™ for fixed terminals using the SIP interface

3. Media gateway – enabling MSC/VLR to transport VoIP calls from the fixed terminals

- At the business premises, IP telephony infrastructure is deployed:

1. Terminals:
    1. IP phones
    2. Normal "POTS" phones with an MTA (Multimedia Terminal Adapter) to enable VoIP connectivity
    3. Personal computers with IP telephony clients (such as Windows Messenger)
2. Existing PBX (note that the PBX functionality can be fully replaced by the solution, in an immediate or gradual manner)
3. A VoIP gateway connecting all non-VoIP terminals and connections (to the PBX, to the PSTN, POTS phone)
4. A Router with firewall and VPN functionality to connect the business to the operator, as well as to the Internet

- Connectivity between the operator and the business can be provided by several methods:

1. Direct broadband landline IP connection (E1, xDSL, Cable, fiber optic, etc.)
2. Fixed Wireless broadband IP Terminal as provided by the mobile operator (if operating a cdma2000 or UMTS network)
3. Fixed Wireless Voice Terminal at the very least